WE CLAIM:

1. A cabin communication system for improving clarity of a voice spoken within an interior cabin having ambient noise, said cabin communication system comprising:

an adaptive speech enhancement filter for receiving an audio signal that includes a first component indicative of the spoken voice, a second component indicative of a feedback echo of the spoken voice and a third component indicative of the ambient noise, said speech enhancement filter filtering the audio signal by removing the third component to provide a filtered audio signal, said speech enhancement filter adapting to the audio signal at a first adaptation rate; and

an adaptive acoustic echo cancellation system for receiving the filtered audio signal and removing the second component in the filtered audio signal to provide an echo-canceled audio signal, said echo cancellation signal adapting to the filtered audio signal at a second adaption rate,

wherein said first adaptation rate and said second adaptation rate are different from each other so that said speech enhancement filter does not adapt in response to operation of said echo-cancellation system and said echo-cancellation system does not adapt in response to operation of said speech enhancement filter.

2. The cabin communication system of claim 1, wherein said first adaptation rate is greater than said second adaptation rate.

3. The cabin communication system of claim 2, wherein said first adaptation rate of said speech enhancement filter is controlled by a step size β, wherein said second adaptation rate of said echo cancellation system is controlled by a step size μ, and wherein β is much less than μ.

4. The cabin communication system of claim 3, wherein said audio signal is sampled at a sampling frequency F_s, wherein n is the number of samples of the audio signal accumulated for block processing by said speech enhancement filter, wherein said echo cancellation system includes a plurality of filters and a variable 1/k is the fraction of said plurality of filters that are updated each sampling period, and wherein:

$$\beta \ll \frac{\mu}{k} \ll \frac{F_s}{n}.$$

5. The cabin communication system of claim 1, wherein said first adaptation rate is an adaptation rate of a long term noise estimate by said speech enhancement filter, said first adaptation rate being much smaller than said second adaptation rate, and said second adaptation rate being much smaller than a basic filter rate of said speech enhancement filter.

6. The cabin communication system of claim 1, further comprising random noise adding means for adding random noise to the filtered audio signal, said echo cancellation system using the filtered audio signal with the random noise added thereto to identify the second component.

7. The cabin communication system of claim 6, wherein the random noise is a dither signal.

8. The cabin communication system of claim 7, wherein the cabin is movable at a variable velocity and the dither signal is scaled to the velocity.

9. A cabin communication system for improving clarity of a voice spoken within an interior cabin having ambient noise, said cabin communication system comprising:

an adaptive speech enhancement filter for receiving an audio signal that includes a first component indicative of the spoken voice, a second component indicative of a feedback echo of the spoken voice and a third component indicative of the ambient noise, said speech enhancement filter filtering the audio signal by removing the third component to provide a filtered audio signal; and

an adaptive acoustic echo cancellation system for receiving the filtered audio signal and removing the second component in the filtered audio signal to provide an echo-canceled audio signal,

wherein said speech enhancement filter and said echo cancellation system are coupled, and

wherein said cabin communication performs a coupled on-line identification of noise and echos in the audio signal to effect closed loop control of the adaptations of said speech enhancement filter and said echo cancellation system.

10. The cabin communication system of claim 9, wherein said speech enhancement filter adapts to the audio signal at a first adaptation rate and said echo cancellation signal adapts to the filtered audio signal at a second adaption rate, and wherein said first adaptation rate and said second adaptation rate are different from each other so that said speech enhancement filter does not adapt in response to operation of said echo-cancellation system and said echo-cancellation system does not adapt in response to operation of said speech enhancement filter.

11. The cabin communication system of claim 10, wherein said first adaptation rate is greater than said second adaptation rate.

12. The cabin communication system of claim 11, wherein said first adaptation rate of said speech enhancement filter is controlled by a step size β, wherein said second adaptation rate of said echo cancellation system is controlled by a step size μ, and wherein β is much less than μ.

13. The cabin communication system of claim 12, wherein said audio signal is sampled at a sampling frequency F_s, wherein n is the number of samples of the audio signal accumulated for block processing by said speech enhancement filter, wherein said echo cancellation system includes a plurality of filters and a variable 1/k is the fraction of said plurality of filters that are updated each sampling period, and wherein:

$$\beta << \frac{\mu}{k} << \frac{F_s}{n}$$

14. The cabin communication system of claim 10, wherein said first adaptation rate is an adaptation rate of a long term noise estimate by said speech enhancement filter, said first adaptation rate being much smaller than said second adaptation rate, and said second adaptation rate being much smaller than a basic filter rate of said speech enhancement filter.

15. The cabin communication system of claim 9, further comprising random noise adding means for adding random noise to the filtered audio signal, said echo cancellation system using the filtered audio signal with the random noise added thereto to identify the second component.

16. The cabin communication system of claim 15, wherein the random noise is a dither signal.

17. The cabin communication system of claim 16, wherein the cabin is movable at a variable velocity and the dither signal is scaled to the velocity.

18. A cabin communication system for improving clarity of a voice spoken within an interior cabin having ambient noise, said cabin communication system comprising:

a microphone for receiving the spoken voice and the ambient noise and for converting the spoken voice and the ambient noise into a first audio signal, the first audio signal having a first component corresponding to the spoken voice and a second component corresponding to the ambient noise;

an adaptive speech enhancement filter for filtering the first audio signal by removing the second component to provide a filtered audio signal, said speech enhancement filter adapting to the first audio signal at a first adaptation rate;

an adaptive acoustic echo cancellation system for receiving the filtered audio signal and providing an echo-canceled audio signal, said echo cancellation signal adapting to the filtered audio signal at a second adaption rate; and

a loudspeaker for converting the echo-canceled audio signal into an output reproduced voice within the cabin including a third component indicative of the first audio signal,

wherein said loudspeaker and said microphone are acoustically coupled so that the output reproduced voice is fed back from said loudspeaker to be received by said microphone and converted with the spoken voice into the first audio signal.

wherein said echo cancellation system removes from the filtered audio signal any portion of the filtered audio signal corresponding to the third component, and

wherein said first adaptation rate and said second adaptation rate are different from each other so that said speech enhancement filter does not adapt in response to operation of said echo-cancellation system and said echo-cancellation system does not adapt in response to operation of said speech enhancement filter.

19. The cabin communication system of claim 18, wherein said first adaptation rate is greater than said second adaptation rate.

20. The cabin communication system of claim 19, wherein said first adaptation rate of said speech enhancement filter is controlled by a step size β, wherein said second adaptation rate of said echo cancellation system is controlled by a step size μ, and wherein β is much less than μ.

21. The cabin communication system of claim 20, wherein said first audio signal is sampled at a sampling frequency F_s, wherein n is the number of samples of the first audio signal accumulated for block processing by said speech enhancement filter, wherein said echo cancellation system includes a plurality of filters and a variable 1/k is the fraction of said plurality of filters that are updated each sampling period, and wherein:

$$\beta \ll \frac{\mu}{k} \ll \frac{F_s}{n}$$

22. The cabin communication system of claim 18, wherein said first adaptation rate is an adaptation rate of a long term noise estimate by said speech enhancement filter, said first adaptation rate is much smaller than said second adaptation rate, and said second adaptation rate is much smaller than a basic filter rate of said speech enhancement filter.

23. The cabin communication system of claim 18, further comprising random noise adding means for adding random noise to the filtered audio signal, said echo cancellation

system using the filtered audio signal with the random noise added thereto to identify the third component.

24. The cabin communication system of claim 23, wherein the random noise is a dither signal.

25. The cabin communication system of claim 24, wherein the cabin is movable at a variable velocity and the dither signal is scaled to the velocity.

26. A method for improving clarity of a voice spoken within an interior cabin having ambient noise, said method comprising the steps of:

adaptively filtering, for speech enhancement, an audio signal that includes a first component indicative of the spoken voice, a second component indicative of a feedback echo of the spoken voice and a third component indicative of the ambient noise, said filtering step removing the third component to provide a filtered audio signal, said filtering step adapting to the audio signal at a first adaptation rate; and

adaptively processing the filtered audio signal to remove the second component by acoustic echo cancellation to provide an echo-cancelled audio signal, said processing step adapting to the filtered audio signal at a second adaption rate,

wherein said first adaptation rate and said second adaptation rate are different from each other so that said filtering step does not adapt in response to operation of said

processing step and said processing step does not adapt in response to operation of said filtering step.

27. The method of claim 26, wherein said first adaptation rate is greater than said second adaptation rate.

28. The method of claim 27, wherein said first adaptation rate of said filtering step is controlled by a step size β, wherein said second adaptation rate of said processing step is controlled by a step size μ, and wherein β is much less than μ.

29. The method of claim 28, wherein the first audio signal is sampled at a sampling frequency F_s, wherein n is the number of samples of the first audio signal accumulated for block processing by said speech enhancement filter, wherein said processing step uses a plurality of filters and a variable 1/k is the fraction of the plurality of filters that are updated each sampling period, and wherein:

$$\beta \ll \frac{\mu}{k} \ll \frac{F_s}{n}$$

30. The method of claim 26, wherein said first adaptation rate is an adaptation rate of a long term noise estimate by said filtering step, said first adaptation rate being much smaller than said second adaptation rate, and said second adaptation rate being much smaller than a basic filter rate of said filtering step.

31. The method of claim 26, further comprising the step of adding random noise to the filtered audio signal, processing step using the filtered audio signal with the random noise added thereto to identify the second component.

32. The method of claim 31, wherein the random noise is a dither signal.

33. The method of claim 32, wherein the cabin is movable at a variable velocity and the dither signal is scaled to the velocity.

34. An adaptive acoustic echo cancellation system for use in a cabin communication system for improving clarity of a voice spoken within an interior cabin, the cabin communication system including a microphone for receiving the spoken voice and for converting the spoken voice and the ambient noise into a first audio signal, the first audio signal having a first component corresponding to the spoken voice, the cabin communication system further including a loudspeaker for outputting a second audio signal within the cabin, wherein the loudspeaker and the microphone are acoustically coupled so that the second audio signal is fed back from the loudspeaker to be received by the microphone and converted with the spoken voice into the first audio signal so that the second audio signal includes a second component indicative of the first component, said echo cancellation system comprising:

calculation means for adaptively calculating a first plurality of current impulse response coefficients of an acoustic transfer function between an output of the loudspeaker and

an input of the microphone based upon an error signal and a second plurality of prior impulse response coefficients of the acoustic transfer function;

acoustic echo filter means for applying the first plurality of current impulse response coefficients to the first audio signal to remove from the first audio signal any portion of the first audio signal corresponding to the second component and to provide an echo-canceled audio signal, the loudspeaker converting the echo-canceled audio signal into the second audio signal; and

error signal calculation means for calculating the error signal by calculating a difference between the first audio signal and the echo-canceled audio signal.

35. The system of claim 34, wherein said acoustic echo filter means comprises a Least Mean Squares filter.

36. The system of claim 34, wherein a number of the first plurality of current impulse response coefficients is limited by a delay in computing the second audio signal.

37. The system of claim 34, further comprising a speech enhancement filter for filtering the first audio signal prior to the first audio signal being supplied as a filtered audio signal to said acoustic echo filter means, and wherein a number of the first plurality of current impulse response coefficients is limited by a sum of a delay in computing the second audio signal and a delay by said speech enhancement filter in filtering the first audio signal.

38. The system of claim 34, further comprising a speech enhancement filter for filtering the first audio signal prior to the first audio signal being supplied as a filtered audio signal to said acoustic echo filter means, wherein the cabin has an acoustic delay in transferring the second audio signal from the loudspeaker to the microphone, and wherein a number of the first plurality of current impulse response coefficients is limited by a sum of a delay in computing the second audio signal, a delay by said speech enhancement filter in filtering the first audio signal and the acoustic delay.

39. A method for adaptive acoustic echo cancellation for use in a cabin communication system for improving clarity of a voice spoken within an interior cabin, the cabin communication system including a microphone for receiving the spoken voice and for converting the spoken voice and the ambient noise into a first audio signal, the first audio signal having a first component corresponding to the spoken voice, the cabin communication system further including a loudspeaker for outputting a second audio signal within the cabin, wherein the loudspeaker and the microphone are acoustically coupled so that the second audio signal is fed back from the loudspeaker to be received by the microphone and converted with the spoken voice into the first audio signal so that the second audio signal includes a second component indicative of the first component, said method comprising the steps of:

adaptively calculating a first plurality of current impulse response coefficients of an acoustic transfer function between an output of the loudspeaker and an input of the microphone based upon an error signal and a second plurality of prior impulse response coefficients of the acoustic transfer function;

applying the first plurality of current impulse response coefficients to the first audio signal for acoustic echo cancellation to remove from the first audio signal any portion of the first audio signal corresponding to the second component and to provide an echo-canceled audio signal, the loudspeaker converting the echo-canceled audio signal into the second audio signal; and

calculating the error signal by calculating a difference between the first audio signal and the echo-canceled audio signal.

40. The method of claim 39, wherein said applying step comprises a Least Mean Squares filtering step.

41. The method of claim 39, wherein a number of the first plurality of current impulse response coefficients is limited by a delay in computing the second audio signal.

42. The method of claim 39, further comprising a speech enhancement filtering step of filtering the first audio signal prior to the first audio signal being supplied as a filtered audio signal to said applying step, and wherein a number of the first plurality of current impulse response coefficients is limited by a sum of a delay in computing the second audio signal and a delay by said speech enhancement filtering step in filtering the first audio signal.

43. The method of claim 39, further comprising a speech enhancement filtering step of filtering the first audio signal prior to the first audio signal being supplied as a

filtered audio signal to said applying step, wherein the cabin has an acoustic delay in transferring the second audio signal from the loudspeaker to the microphone, and wherein a number of the first plurality of current impulse response coefficients is limited by a sum of a delay in computing the second audio signal, a delay by said speech enhancement filtering step in filtering the first audio signal and the acoustic delay.

44. A movable vehicle cabin having ambient noise, said cabin comprising:

means for causing movement of said cabin, wherein at least a portion of the ambient noise during movement is a result of the movement; and

a cabin communication system for improving clarity of a voice spoken within an interior of said cabin, wherein said cabin communication system comprises:

a microphone for receiving the spoken voice and the ambient noise and for converting the spoken voice and the ambient noise into a first audio signal, the first audio signal having a first component corresponding to the spoken voice and a second component corresponding to the ambient noise;

an adaptive speech enhancement filter for filtering the first audio signal by removing the second component to provide a filtered audio signal, said speech enhancement filter adapting to the first audio signal at a first adaptation rate;

an adaptive acoustic echo cancellation system for receiving the filtered audio signal and providing an echo-canceled audio signal, said echo cancellation signal adapting to the filtered audio signal at a second adaption rate; and

a loudspeaker for converting the echo-canceled audio signal into an output reproduced voice within the cabin including a third component indicative of the first audio signal,

wherein said loudspeaker and said microphone are acoustically coupled so that the output reproduced voice is fed back from said loudspeaker to be received by said microphone and converted with the spoken voice into the first audio signal.

wherein said echo cancellation system removes from the filtered audio signal any portion of the filtered-audio signal corresponding to the third component, and

wherein said first adaptation rate and said second adaptation rate are different from each other so that said speech enhancement filter does not adapt in response to operation of said echo-cancellation system and said echo-cancellation system does not adapt in response to operation of said speech enhancement filter.

45. The cabin of claim 44, wherein said first adaptation rate is greater than said second adaptation rate.

46. The cabin of claim 45, wherein said first adaptation rate of said speech enhancement filter is controlled by a step size β, wherein said second adaptation rate of said echo cancellation system is controlled by a step size μ, and wherein β is much less than μ.

47. The cabin of claim 46, wherein said first audio signal is sampled at a sampling frequency F_s, wherein n is the number of samples of the first audio signal accumulated

for block processing by said speech enhancement filter, wherein said echo cancellation system includes a plurality of filters and a variable 1/k is the fraction of said plurality of filters that are updated each sampling period, and wherein:

$$\beta << \frac{\mu}{k} << \frac{F_s}{n}.$$

48. The cabin of claim 44, wherein said first adaptation rate is an adaptation rate of a long term noise estimate by said speech enhancement filter, said first adaptation rate is much smaller than said second adaptation rate, and said second adaptation rate is much smaller than a basic filter rate of said speech enhancement filter.

49. The cabin of claim 44, further comprising random noise adding means for adding random noise to the filtered audio signal, said echo cancellation system using the filtered audio signal with the random noise added thereto to identify the third component.

50. The cabin of claim 49, wherein the random noise is a dither signal.

51. The cabin of claim 50, wherein the cabin is movable at a variable velocity and the dither signal is scaled to the velocity.